WHAT IS CLAIMED IS:

1. A hearing aid, comprising:

an input signal channel providing digital input signals;

a signal path adapted to process said digital input signals in accordance with a predetermined signal processing algorithm to produce a digital output signal, wherein said signal path further comprises at least one signal processing function operating on a warped frequency scale; and

an output conversion means adapted to convert said output signals to an audio output.

2. The hearing aid of claim 1, wherein said at least one signal processing function further comprises a plurality of cascaded all-pass filters.

3. The hearing aid of claim 1, wherein said warped frequency scale approximates a Bark scale.

~~4. A dynamic range compressor, comprising:~~

an input signal channel providing digital input signals;

a plurality of cascaded all-pass filters, wherein said digital input signals pass through said plurality of cascaded all-pass filters, and wherein said plurality of cascaded all-pass filters output a sequence of delayed samples;

means for applying a frequency domain transform on said sequence of delayed samples, wherein a warped sequence results from said frequency domain transform applying means;

means for calculating a plurality of frequency domain level estimates from said warped sequence;

means for calculating a plurality of frequency domain gain coefficients from said plurality of frequency domain level estimates;

means for applying an inverse frequency domain transform on said plurality of frequency domain gain coefficients, wherein a set of compression filter coefficients of a compression gain filter result from said inverse frequency domain transform applying means; and

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means for convolving said sequence of delayed samples with said set of compression filter coefficients to produce a digital output signal.

5. The dynamic range compressor of claim 4, further comprising a hearing aid, wherein the dynamic range compressor is incorporated within said hearing aid.

6. The dynamic range compressor of claim 4, wherein said plurality of frequency domain gain coefficients comprise a warped time-domain filter.

7. The dynamic range compressor of claim 4, further comprising means for windowing said sequence of delayed samples, wherein a windowed sequence of delayed samples results from said windowing means, and wherein said warped sequence results from applying said frequency domain transform to said windowed sequence of delayed samples.

8. The dynamic range compressor of claim 4, further comprising a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals.

9. The dynamic range compressor of claim 8, further comprising an output transducer, said output transducer converting said analog output signals to an audio output.

10. The dynamic range compressor of claim 4, said plurality of cascaded all-pass filters comprising a plurality of first order all-pass filters.

11. The dynamic range compressor of claim 4, said sequence of delayed samples comprising 16 samples.

12. The dynamic range compressor of claim 4, further comprising a digital processor, wherein said digital processor is adapted to provide said frequency domain transform applying means, said frequency domain level estimates calculating means, said frequency domain gain coefficients calculating means, said inverse frequency domain transform applying means, and said means for convolving said sequence of delayed samples.

13. The dynamic range compressor of claim 12, wherein said digital processor comprises a software programmable digital signal processor.

14. The dynamic range compressor of claim 4, wherein said frequency domain transform applying means uses a transform selected from the group consisting of discrete Fourier transforms, fast Fourier transforms, Goertzel transforms, and discrete cosine transforms.

15. The dynamic range compressor of claim 4, further comprising:

an input transducer, said input transducer converting audio input signals to analog input signals; and

an analog-to-digital converter, said analog-to-digital converter converting said analog input signals to said digital input signals.

16. The dynamic range compressor of claim 4, further comprising:

a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals; and

an output transducer, said output transducer converting said analog output signals to an audio output.

17. A dynamic range compressor, comprising:

an input signal channel providing digital input signals;

an input data buffer, said input data buffer holding at least one block of data comprised of a portion of said digital input signals;

a plurality of cascaded all-pass filters, wherein a first block of said digital input signals pass from said input data buffer through said plurality of cascaded all-pass filters, and wherein said plurality of cascaded all-pass filters output a first sequence of delayed samples;

means for windowing a first portion of said first sequence of delayed samples, wherein a first windowed sequence of delayed samples results from said windowing means;

means for applying a first frequency domain transform on said first windowed sequence of delayed samples, wherein a first warped sequence results from said first frequency domain transform applying means;

means for calculating a first plurality of frequency domain level estimates of said first warped sequence;

means for windowing a second portion of said first sequence of delayed samples, wherein a second windowed sequence of delayed samples results from said windowing means;

means for applying a second frequency domain transform on said second windowed sequence of delayed samples, wherein a second warped sequence results from said second frequency domain transform applying means;

means for calculating a second plurality of frequency domain level estimates of said second warped sequence;

means for summing said first and second plurality of frequency domain level estimates, wherein a summed first and second plurality of frequency domain level estimates results from said summing means;

means for normalizing said summed first and second plurality of frequency domain level estimates, wherein a normalized first and second plurality of frequency domain level estimates results from said normalizing means;

means for calculating a plurality of frequency domain gain coefficients from said normalized first and second plurality of frequency domain level estimates;

means for applying an inverse frequency domain transform on said plurality of frequency domain gain coefficients, wherein a set of compression filter coefficients of a compression gain filter result from said inverse frequency domain transform applying means;

means for convolving a second sequence of delayed samples with said compression filter coefficients, said second sequence of delayed samples produced by a second block of said digital input signals passing from said input data buffer through said plurality of cascaded all-pass filters, wherein a digital output signal results from said convolving means.

18. The dynamic range compressor of claim 17, further comprising a hearing aid, wherein the dynamic range compressor is incorporated within said hearing aid.

19. The dynamic range compressor of claim 17, wherein said plurality of frequency domain gain coefficients comprise a warped time-domain filter.

20. The dynamic range compressor of claim 17, further comprising a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals.

21. The dynamic range compressor of claim 20, further comprising an output transducer, said output transducer converting said analog output signals to an audio output.

22. The dynamic range compressor of claim 17, said plurality of cascaded all-pass filters comprising a plurality of first order all-pass filters.

23. The dynamic range compressor of claim 17, further comprising a digital processor, wherein said digital processor is adapted to provide said windowing means, said means for applying said first and second frequency domain transforms, said means for calculating said first and second plurality of frequency domain level estimates, said summing means, said normalizing means, said frequency domain gain coefficients calculating means, said inverse frequency domain transform applying means, and said convolving means.

24. The dynamic range compressor of claim 17, wherein said means for applying said first and second frequency domain transforms use a transform selected from the group consisting of discrete Fourier transforms, fast Fourier transforms, Goertzel transforms, and discrete cosine transforms.

25. The dynamic range compressor of claim 17, further comprising:

an input transducer, said input transducer converting audio input signals to analog input signals; and

an analog-to-digital converter, said analog-to-digital converter converting said analog input signals to said digital input signals.

26. The dynamic range compressor of claim 17, further comprising:

a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals; and

an output transducer, said output transducer converting said analog output signals to an audio output.

27. The hearing aid of claim 17, wherein said windowing means provides a 50 percent overlap of said first and second pluralities of frequency domain level estimates.

28. The hearing aid of claim 17, wherein a quantity of samples corresponding to said first block of said digital input signals is equivalent to a quantity of first order all-pass filters corresponding to said plurality of cascaded all-pass filters.

29. The hearing aid of claim 28, wherein said first portion of said first sequence of delayed samples is comprised of a first half of said first sequence of delayed samples and said second portion of said first sequence of delayed samples is comprised of a second half of said first sequence of delayed samples.

30. A hearing aid, comprising:

an input signal channel providing digital input signals;

an input data buffer, said input data buffer holding a block of data of size M comprised of a portion of said digital input signals;

a plurality of cascaded all-pass filters comprised of 2M cascaded all-pass filters, wherein a first block of said digital input signals pass from said input data buffer through said plurality of cascaded all-pass filters to form a first sequence of delayed samples and wherein a second block of said digital input signals pass from said input data buffer through said plurality of cascaded all-pass filters to form a second sequence of delayed samples, and wherein said first sequence of delayed samples and said second sequence of delayed samples form a combined sequence of delayed samples;

means for windowing a first portion of said combined sequence of delayed samples, wherein said first portion is of size M, wherein a windowed sequence of delayed samples results from said windowing means;

means for applying a 2M-point frequency domain transform on said windowed sequence of delayed samples, wherein a warped sequence results from said frequency domain transform applying means;

means for calculating a plurality of frequency domain level estimates of said warped sequence;

means for calculating a plurality of frequency domain gain coefficients from said plurality of frequency domain level estimates;

means for applying an inverse frequency domain transform on said plurality of frequency domain gain coefficients, wherein a set of compression filter coefficients of a compression gain filter result from said inverse frequency domain transform applying means; and

means for convolving a second portion of said combined sequence of delayed samples with said compression filter coefficients, wherein said second portion is of size M, wherein a digital output signal results from said convolving means.

31. The dynamic range compressor of claim 30, further comprising a hearing aid, wherein the dynamic range compressor is incorporated within said hearing aid.

32. The dynamic range compressor of claim 30, wherein said plurality of frequency domain gain coefficients comprise a warped time-domain filter.

33. The dynamic range compressor of claim 30, further comprising a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals.

34. The dynamic range compressor of claim 33, further comprising an output transducer, said output transducer converting said analog output signals to an audio output.

35. The dynamic range compressor of claim 30, said plurality of cascaded all-pass filters comprising a plurality of first order all-pass filters.

36. The dynamic range compressor of claim 30, further comprising a digital processor, wherein said digital processor is adapted to provide said windowing means, said means for applying said 2M-point frequency domain transform, said means for calculating said plurality of frequency domain level estimates, said frequency domain gain coefficients calculating means, said inverse frequency domain transform applying means, and said convolving means.

37. The dynamic range compressor of claim 30, wherein said means for applying said frequency domain transform uses a transform selected from the group

consisting of discrete Fourier transforms, fast Fourier transforms, Goertzel transforms, and discrete cosine transforms.

38. The dynamic range compressor of claim 30, further comprising:

an input transducer, said input transducer converting audio input signals to analog input signals; and

an analog-to-digital converter, said analog-to-digital converter converting said analog input signals to said digital input signals.

39. The dynamic range compressor of claim 30, further comprising:

a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals; and

an output transducer, said output transducer converting said analog output signals to an audio output.

40. A method of processing sound in a hearing aid, comprising the steps of:

receiving digital input signals;

passing a portion of said digital input signals through a plurality of cascaded all-pass filters to form a sequence of delayed samples;

windowing said sequence of delayed samples;

applying a frequency domain transform to said windowed sequence of delayed samples to form a warped sequence;

calculating a plurality of frequency domain level estimates from said warped sequence;

calculating a plurality of frequency domain gain coefficients from said plurality of frequency domain level estimates to form a warped time domain filter;

applying an inverse frequency domain transform on said plurality of frequency domain gain coefficients to form a set of compression filter coefficients; and

convolving said sequence of delayed samples with said compression filter coefficients to form a digital output signal.